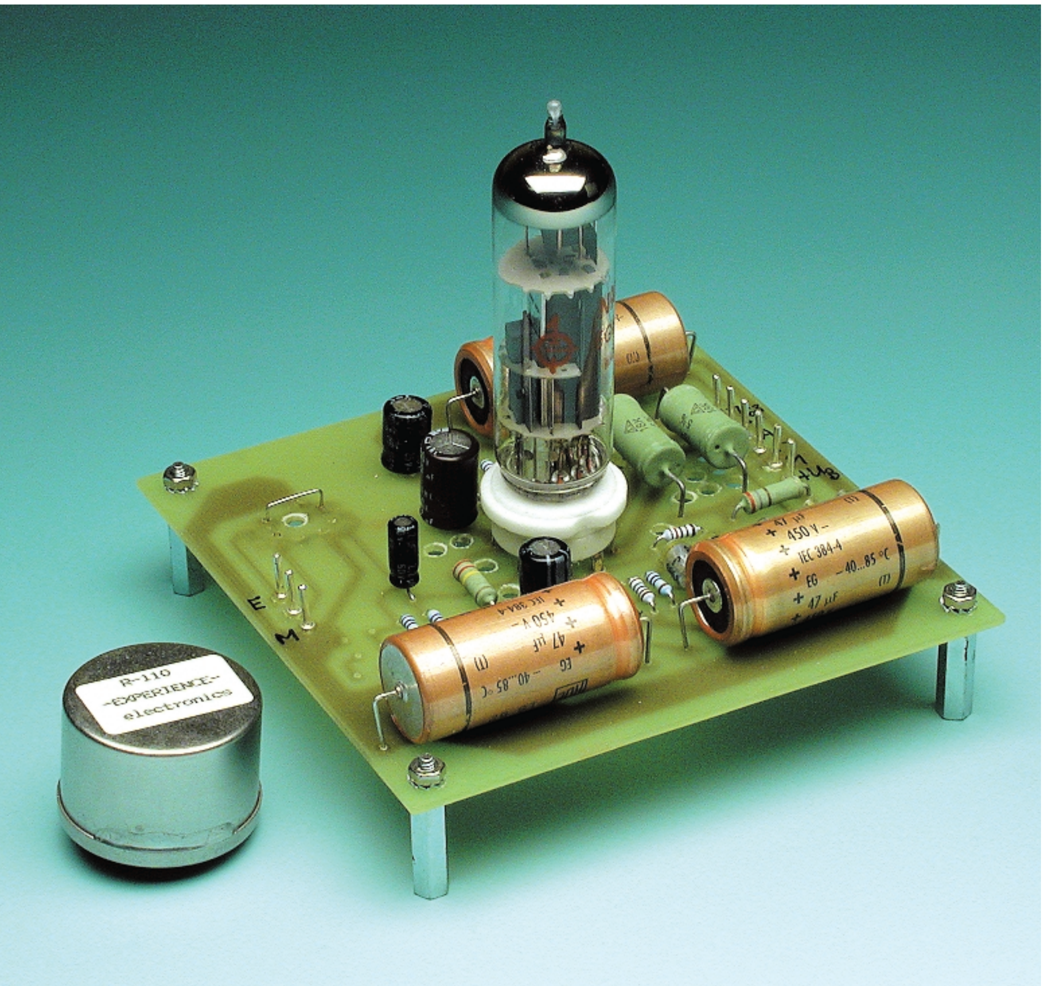
Valved RIAA Preamplifier

using an ECL86

Design by G. Haas

In spite of CDs and digital audio signal processing, there are still people who are devoted to the vinyl phonograph record. And they enjoy their analogue treasures the most in the company of contemporary valve technology.



For nearly a century, vinyl phonograph records played an important role as a storage medium for voice and music. Although nowadays the CD has taken over the leading role, analogue records are still widely distributed. Hundreds of millions of records are present in archives and private record cabinets. These include irreplaceable documentary recordings, collectors' items and many commonplace recordings that reflect the spirit of a particular era and the technology available at that time, as well as recordings that are technically and artistically outstanding. For friends of analogue recordings, there is still a market for phonograph records, in which re-pressings or new pressings can be obtained. Due to the small quantities produced, as well as the high quality, the prices are fairly high.

In a direct comparison on a good stereo installation, a high-quality phonograph record can easily match or surpass the quality of a CD. The only disadvantages of records are the mechanical noise from the pick-up needle and the crackling caused by dust and electrostatic discharges. Consequently,

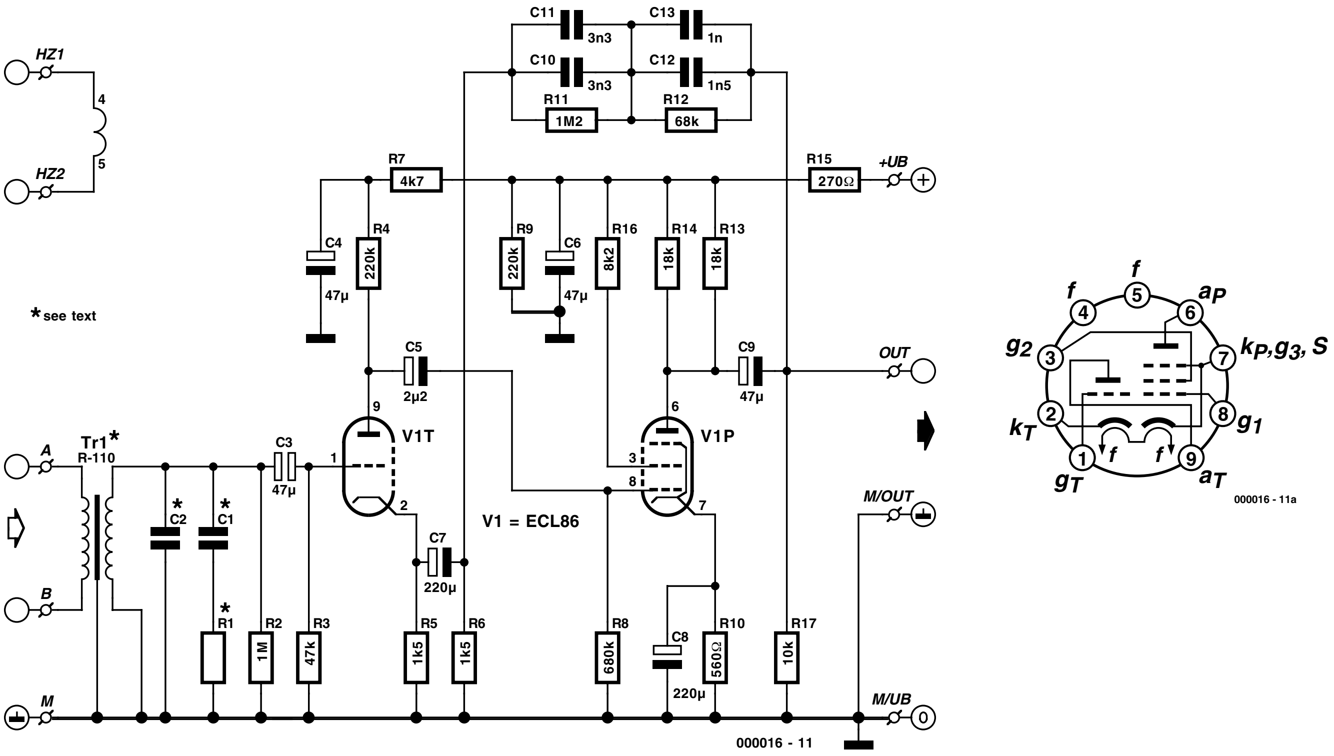


Figure 1. Schematic diagram of the RIAA preamplifier with the optional input transformer.

there is a steady demand for good equalising preamplifiers, including those based on valve technology. This is where the circle closes. Before the coming of semiconductor technology, valve technology was used for everything. This put its stamp on the recording technology and the 'sound'. If we revert to using valves to amplify the phonograph signal, we come closer to the sound experience of earlier times. That is the *raison d'être* for this construction project.

A valved opamp

An equalising preamplifier using valves cannot be compared to one built with semiconductors. Modern opamps have very high open-loop gains, so there is lots of reserve gain available for negative feedback. It is very costly and difficult to achieve equally high levels of open-loop gain with a valve amplifier.

Dual triodes or audio-frequency input-stage pentodes are usually used in these designs. However, here again we chose to take a somewhat unconventional approach and use an ECL86, as can be seen from the schematic diagram of the preamplifier in **Figure 1**. A high-gain audio triode combined with a high-current audio output pentode can be regarded as a valve opamp (high gain, high input impedance and low output impedance). In order to better understand the operation of this circuit, let's first look at its functions in detail.

If the negative feedback network formed by R11, R12 and C10–C13 is disconnected, the open-loop gains of the valves are determined by their associated circuitry. The cathode resistor of the pentode is bridged by a large electrolytic capacitor. This means that the gain of the pentode is equal to $A_v = R_a \times G_m$ (see the box 'Gain Calculations'). With the triode, the DC operating point is determined by R5. Capacitor C7 represents a short circuit for AC voltages, so resistors R5 and R6 are effectively connected in parallel. Due to negative feedback applied via the cathode resistor, the triode will not achieve its full theoretical gain level. Without this negative feedback, the triode would have a gain of 77 in this configuration, while the pentode would have a gain of 90. The combined gain would be 6930, which is around 76.5 dB. The actual gain of the triode is significantly less than 77, due to the negative feedback. In addition, we have to take into account the range of variation in the characteristics of individual valves. Consequently, the basic gain is set to around 35 dB at 1 kHz. This is not breathtakingly high, but it has the advantage that you do not have to use a selected valve, but instead can use any one you wish.

It is not necessary to describe the schematic in more detail, since the functions of the individual components are evident. The operating principle of the ECL86 in this configuration has been thoroughly described in the Valve Preamplifier article. The quality of the components must match what is stated in the components list. If you adhere to this rule, you will have no trouble achieving performance figures equal to those given in the technical specifications.

With this circuit, the old debate about electrolytic capacitors in the signal path will flare up once again. The main advantage of electrolytic capacitors is that they provide very high capacitance in a very small space. This makes it possible to have low-impedance signal coupling, even at very low frequencies. In addition, a lot of technical progress has been made in the construction of electrolytic capacitors in the last few years — more than

Technical Specifications

Stabilised operating voltage		330 V
Nominal output voltage	into 100 kΩ	200 mV
Output impedance	at 20 Hz	2 kΩ
	at 1 kHz	150 Ω
	at 20 kHz	25 Ω
Current consumption	approximately	20 mA

MM (source impedance 750 Ω)

THD+N	BW= 80 kHz, 1 kHz	< 0.06 %
THD+N	A-weighted	< 0.014 %
S/N	22 Hz – 22 kHz	> 65 dB
S/N	A-weighted	> 76 dB
Gain	1 kHz, U_N = 3.5 mV	35 dB

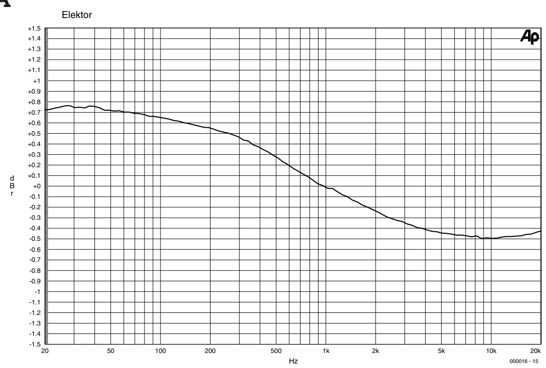
MC (source impedance 25 Ω, using an R-110 input transformer)

THD+N	BW = 80 kHz, 1 kHz	< 0.07 %
THD+N	A-weighted	< 0.018 %
S/N	22 Hz – 22 kHz	> 63 dB
S/N	A-weighted	> 74 dB
Gain	1 kHz, U_{IN} = 0.37 mV	55 dB

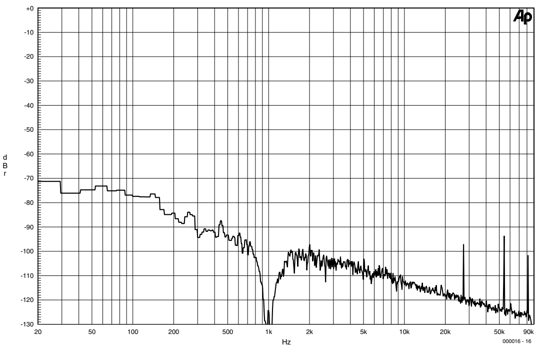
Figure A shows the deviation from the RIAA curve. The relative deviation remains within +0.8 dB / –0.5 dB with a load of 47 kΩ. Due to the high output impedance, departures from these results may occur with other loads, particularly at low frequencies.

Figure B shows the frequency spectrum for a 200-mV, 1-kHz signal with a 100-kΩ load. It can be seen that the THD+N value consists almost exclusively of the noise component. The peaks at 25 kHz and above are generated by the switching power supply (soon to appear in Elektor Electronics!). Lying at –90 dB, they are not only so low as to have no effect on the measurement, they are also far outside the range of human hearing.

A



B



with other types of passive components. We suggest that you first build the circuit and listen to the results. After this, you are free to experiment with other types of capacitors.

MM or MC?

Now we come to an interesting circuit detail at the input. If a conventional MM (moving-magnet) cartridge is used, it is connected directly to C3. Resistor R3 then provides the standard termination impedance of 47 kΩ. Resistor R2 functions only as a bleeder resistor for DC voltages, and C2 can be used as needed. A particular capacitive load is prescribed for each type of MM cartridge, in order to obtain a linear frequency response. Usually, the connecting cable of the phonograph is dimensioned so that it provides the correct load in combination with the stray circuit capacitance. If this is not sufficient, C2 must be used to provide compensation. A ceramic capacitor is usually used here, with a value ranging between 10 pF and several hundred picofarads.

If an MC (moving-coil) cartridge is to be used, it is both helpful and worthwhile to employ an audio input transformer. Moving-coil cartridges have significantly better reproduction characteristics than moving-magnet cartridges, due to their operating principle. Their disadvantage is that the output voltage is approximately a factor of ten lower. This can be compensated by using an R-110 input transformer, which is a toroidal-core transformer enclosed in Mu-metal. It boosts the low signal voltage by a factor of 10 (20 dB), with practically no noise and a very small distortion component. This type of transformer can only produce odd harmonics. The distortion factor depends on the primary voltage level and the frequency. The lower the frequency and the higher the voltage, the higher the distortion factor. The characteristics of the R-110 transformer are measured at 1 mV, which is a good standard value for MC cartridges.

The Mu-metal enclosure for the transformer is relatively expensive, but it is absolutely necessary. Without it, all induced magnetic disturbances would be correspondingly amplified along with the signal and would degrade the signal-to-noise ratio. Using the transformer allows us to avoid trying to get the utmost out of amplifier technology, with all the associated disadvantages. In order for the input transformer to work perfectly, certain basic considerations must be taken into account. The transformer used here changes the voltage by a factor t = 10. The voltage is transformed upwards by a factor of 10, but the impedance is transformed by a factor equal to t^2. If the transformer is terminated with 47 kΩ, for example, the termination impedance R seen by the moving-coil cartridge is (R3 / t^2) = (47 kΩ / 10^2) = 470 Ω.

In order for this low impedance to have an effect at the amplifier input, C3 must have a high capacitance. The 3-dB corner frequency lies at around 0.07 Hz in this case. In addition, the amplifier input sees the low source impedance, which has a beneficial effect on the noise behaviour. The transformer

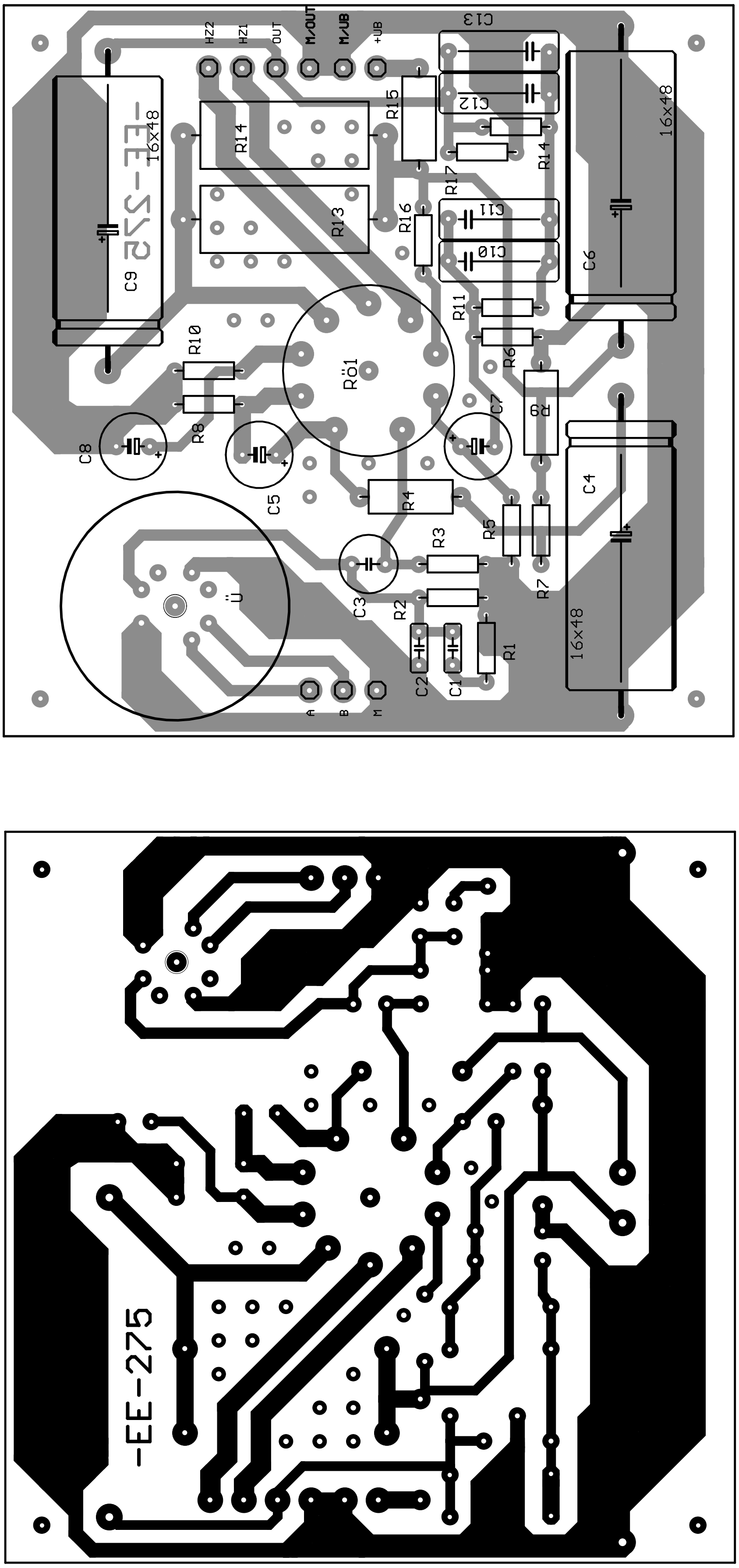


Figure 2. The printed circuit board, which supports one (monophonic) channel, is available from the author.

has a screen between the primary and secondary windings, which diverts interference to ground.

In order to use a moving-coil cartridge correctly, you must observe the specifications given on its data sheet. For example, if a termination resistance of 1 kΩ is specified, R3 must be increased to 100 kΩ. In this way, the cartridge sees the required termination resistance, while the source resistance seen by the preamplifier is the internal resistance of the moving-coil cartridge. R2 can be omitted if an input transformer is used, since the transformer winding provides the DC path. The frequency response can be corrected using C2, C1 and R1. C2 represents the nec-

COMPONENTS LIST

Resistors:

(metal film, 0.7 W, 1% tolerance,
MO = 5%, tolerance)
R1 = see text
R2 = 1MΩ (see text)
R3 = 47kΩ
R4 = 220kΩ, MO, 2W
R5,R6 = 1kΩ5
R7 = 4kΩ7
R8 = 680kΩ
R9 = 220kΩ, MO, 2 W
R10 = 560Ω
R11 = 1MΩ2
R12 = 68kΩ
R13,R14 = 18kΩ, MO, 4.5 W
R15 = 270Ω, MO, 2W
R16 = 8kΩ2
R17 = 10kΩ

Capacitors:

C1,C2 = see text
C3 = 47µF 35V bipolar
C4 = 47µF 450 V axial
C5 = 2µF2 400V, lead pitch 5mm
C6 = 47µF 450V axial
C7,C8 = 220µF 25V, lead pitch 5mm
C9 = 47µF 450V axial
C10,C11 = 3nF3, 2.5 % polypropylene, min. 100V
C12 = 1nF5, 2.5 %, polypropylene, min. 100V
C13 = 1nF, 2.5 %, polypropylene, min. 100V

Miscellaneous:

V1 (Rö1) = ECL86
1 noval (9-pin) socket, ceramic, for board mounting
1 PCB, epoxy resin strengthened, 70µm copper layer
1 Moving-Coil transformer, type R-110, see text
Solder pins

essary capacitive load, which depends on the cartridge employed and the stray circuit capacitance. Depending on the configuration, there may be a resonance peak in the audible range. This can be suppressed using R1 and C1, in order to make the frequency response linear. If C1 and R1 are needed, their values will lie in the range of 22 pF to 1 nF and 5 kΩ to 20 kΩ.

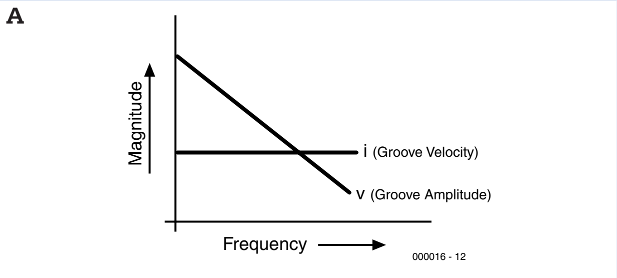
Construction tips

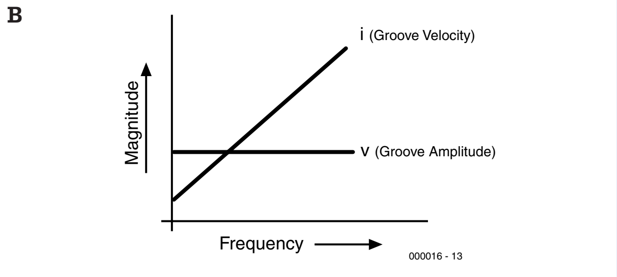
The MC cartridge should be connected as symmetrically as possible using XLR connectors. This is the only way to conduct the very small pickup signal to the amplifier input without introducing interference. Connect the screen braid to pin 1 in the XLR connector. The following small table shows how everything is arranged:

RIAA Cutting

RIAA is an acronym that stands for 'Record Industry Association of America'. This organisation has specified how phonograph records are to be cut (mastered) and reproduced, so that every record can be played on every phonograph anywhere in the world. What is the purpose of the well-known RIAA curve?

When a record is recorded, the first thing that has to be done is to decide on the groove spacing. The narrower the spacing, the more program material that can be placed on each side. The price of narrow spacing is reduced lateral excursion, which means reduced dynamic range. With a given groove width, the maximum excursion must be limited, since otherwise groove overcutting will occur. This is especially true in the low frequency range (below around 500 Hz), where the largest amplitudes appear.





A phonograph record is cut using an electrodynamic process. The signal currents for the individual channels are passed through separate coils that drive the cutting stylus, similar to how a dynamic loudspeaker is driven. A spring/mass system is driven by a coil/magnet system. The driven cutting stylus is free to move along only two axes. The impedance of the coils is frequency-dependent. If constant-amplitude drive is used, the cutting velocity rises with increasing frequency (see **Figure A**). With constant-cutting-velocity drive, the amplitude drops with increasing frequency (see **Figure B**.) Since the frequency range that must be handled runs from 20 Hz to 20 kHz, the amplitude ratio with constant cutting velocity is 1:1000 (60 dB).

Such an extremely large dynamic ratio would mean that no usable signal-to-noise ratio would be left at high frequencies. For the optimal utilisation of the surface area of the record, constant-amplitude recording is ideal, but this means that the cutting velocity rises with increasing frequency. When the record is played back using a coil/magnet system, the law of induction thus says that the output voltage will rise with increasing frequency.

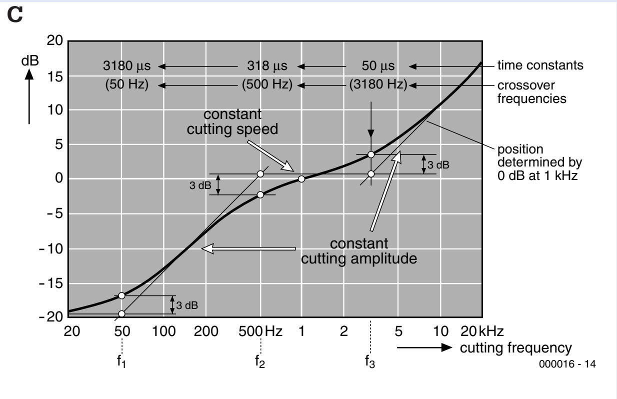
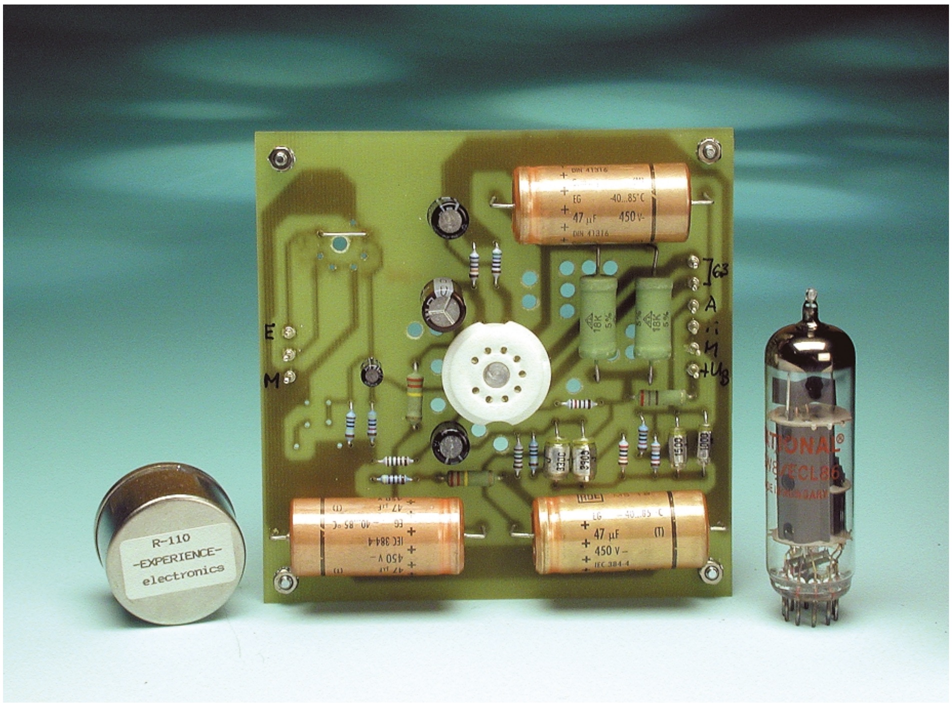


Figure C shows the RIAA recording curve with its corner frequencies and associated time constants. Constant-amplitude recording is used in the frequency range between f1 and f2, in order to limit the maximum excursion and thus avoid groove overcutting. This can be considered to be the same as bass attenuation. In the region between f2 and f3, the behaviour of the coil/magnet system is taken into account and constant-velocity recording is used. Above f3, up to the end of the range, constant-amplitude recording is again used. The net result is that the bass frequencies are attenuated enough to avoid groove overcutting, while the surface area of the record is efficiently exploited. The mid-range frequencies are handled neutrally in comparison to the low frequencies, while the high frequencies are emphasised, which leads to a significant improvement in the signal-to-noise ratio. In the frequency range from approximately f3 upwards, the human ear has increased sensitivity to noise components. During playback, the high frequencies are reproduced too loud, so they must be attenuated. This is precisely the effect that improves the signal-to-noise ratio. The bass frequencies, by contrast, must be emphasised, which leads to increased sensitivity to induced mains-frequency signals and their harmonics. If an equalising preamplifier can maintain the complementary reproduction curve within less than ±1 dB, it is considered to be high-end equipment. The numerical values in the table, which represent the RIAA curve with reference to 1 kHz = 0 dB, are a useful aid for making measurements.

Hz	dB	Hz	dB
20	+19.3	800	+0.7
30	+18.6	1k	0.0
40	+17.8	1.5k	–1.4
50	+17.0	2k	–2.6
60	+16.1	3k	–4.8
80	+14.5	4k	–6.6
100	+13.1	5k	–8.2
150	+10.3	6k	–9.6
200	+8.2	8k	–11.9
300	+5.5	10k	–13.7
400	+3.8	15k	–17.2
500	+2.6	20k	–19.6

Standard lead colour	XLR pin	RIAA-Preamp
white	2	a left
blue	3	b left
screen	1	m left
red	2	a right
green	3	b right
screen	1	m right



If you do not use a moving-coil input transformer, you should use a quasi-symmetric lead arrangement with Cinch connectors, instead of the usual asymmetric arrangement. Connect the screen and the blue or green lead together inside each Cinch connector. This keeps the screens free of signal voltages in this case as well.

The amplifier circuit is relatively simple, but it has very good basic characteristics. Very high-quality construction is absolutely necessary if these are to be fully realised. The amplifier circuit is designed to be monaural, so separate printed circuit boards (as shown in **Figure 2**) are needed for the two channels. The circuit board is not available from Readers Services, but it can be obtained from the author. Mount the two circuit boards in a generously sized, screened enclosure, well separated from interference sources. A high degree of channel separation can be maintained by spacing the two boards widely apart, or by placing a sheet-metal screen between the two boards if space is tight. This makes a lot of difference to the sound. The standard requires >26 dB at 1 kHz, which lies within the realm of what is possible.

The power supply does not belong in the amplifier enclosure. Only the amplifier boards are mounted in the non-magnetic metallic enclosure, while the power supply is housed externally and placed at a sufficient remove from the equalising amplifier. The well-filtered, floating supply voltages (filament and high voltage) are fed to the amplifier enclosure via separate cables. This avoids the superimposition of the filament and high-voltage currents, which could increase the base noise level. The input and output sockets of the amplifier are mounted isolated from ground. The negative poles of the high voltage and filament supplies are connected together, along with the enclosure and signal ground, at a single point. This results in a non-grounded power supply and avoids hum loops. If the signal leads in the phonograph are isolated from the chassis, the phonograph chassis must be connected to the amplifier enclosure via the lead provided for this purpose (black).

The ECL86 has a filament voltage of 6.3 V at 0.66 A. The filaments must be connected in series to match the DC filament voltage of 12.6 V.

Once you have carefully built this preamplifier, you are ready to enjoy unimpeded listening pleasure. Even old monophonic records will sound distinctly better, since a stereo cartridge has significantly better tracking characteristics than a mono cartridge. The value of your record collection will be distinctly enhanced by this low-noise, low-distortion amplifier using primarily K2 components.

(000016-1)

Additional information is available from:

Experience Electronics
Mr. Gerhard Haas
Weststrasse 1
D-89542 Herbrechingen
Germany.
Tel.: (+49) 7325 5318
Fax: (+49) 7324 2553
e-mail:
experience.electronics@t-online.de

Gain Calculations

The gain of a triode is determined by its transconductance, its internal resistance and the value of the anode resistor that is used. The gain formula is $A_v = R_a \times g_m$, where g_m is the dynamic transconductance. The values of the static transconductance and internal resistance can be obtained from the valve data book. The value of the dynamic transconductance, which ultimately determines the gain of the triode circuit, must be calculated using the formula

$$g_m = G_m \cdot \frac{R_i}{R_i + R_a}$$

where

g_m = dynamic transconductance
G_m = static transconductance
R_i = internal resistance
R_a = anode resistance

For pentodes, the formula is $A_v = R_a \times G_m$. A tolerance of 5% for the anode resistor is considered to be very precise, since many parameters of active components often shows a wide range of variation. For example, if the data book gives a value of 1.6 mA/V for the transconductance of an ECC83, a variation of ±30% is easily possible. The internal resistance amounts to 62.5 kΩ. You can easily calculate how much the actual gain can vary for a given value of the anode resistor. The ECL86 consists of half of an ECC83 together with an audio output pentode, which has a transconductance of 10 mA/V.